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BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

Application Number: 10/803,420
Filing Date: March 18, 2004
Appellant(s): SINGHAL ET AL.

01/12/2010

SINGHAL ET AL.
For Appellant

EXAMINER'S ANSWER

This is in response to the appeal brief filed 11/04/2009 appealing from the Office action mailed 06/04/2009.

(1) Real Party in Interest

A statement identifying by name the real party in interest is contained in the brief.

(2) Related Appeals and Interferences

The examiner is not aware of any related appeals, interferences, or judicial proceedings which will directly affect or be directly affected by or have a bearing on the Board's decision in the pending appeal.

(3) Status of Claims

The statement of the status of claims contained in the brief is correct.

(4) Status of Amendments After Final

No amendment after final has been filed.

(5) Summary of Claimed Subject Matter

The summary of claimed subject matter contained in the brief is correct.

(6) Grounds of Rejection to be Reviewed on Appeal

The appellant's statement of the grounds of rejection to be reviewed on appeal is correct.

(7) Claims Appendix

The copy of the appealed claims contained in the Appendix to the brief is correct.

(8) Evidence Relied Upon

US 5781696	Oh et al.	07-1998
US 6915263	Chen et al.	07-2005
US 5684829	Kizuki et al.	11-1997

(9) Grounds of Rejection

The following ground(s) of rejection are applicable to the appealed claims:

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the

> invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1, 4-6, 9-11, and 14-18 are rejected under 35 U.S.C. 103(a) as being unpatentable over Oh et al. US 5781696 (hereinafter Oh) in view of Chen et al. US 6915263 B1 (hereinafter Chen).

Re claims 1, 6, and 11, Oh teaches a method for speeding up an encoded original audio signal, said original audio signal having an original frequency and original playback speed, said method comprising:

retrieving frames of the original audio signal (Fig. 5);

wherein said desired playback speed is greater than the original playback speed (col. 5 lines 60-65);

applying a window function (col. 5 line 65 – col. 6 line 2) to the remaining frames

converting the signal with the windowed frames from digital to analog format;

using the original frequency to playback the analog format signal (col. 6 lines 38-46)

skipping frames at a rate according to a desired playback speed (Col. 1 lines 33-45, every other frame at a higher play back speed);

However, Oh fails to teach receiving the encoded original audio signal;

applying a window function (col. 5 line 65 – col. 6 line 2) to the remaining frames

Chen teaches error reduction of encoded frames, wherein Chen teaches error entries of error array 370 can be computed and stored by the parser process 270 of the decoder 200. There are several ways in which the AC3 data can indicate that errors are

contained within a frame of encoded data. In one method, the decoder 200 can be informed of the error frame by the transport system which delivers the data. The data integrity can also be checked using the embedded CRC 220 fields for each encoded frame. Methods for using the CRC fields of an encoded frame for error detection are well known. Also, well known consistency checks on the received bitstream 134 can also be used to indicate that errors are present in a particular encoded frame. It is appreciated that at step 305 of FIG. 4, any of a number of well known processes can be used for generating the error array 370 of FIG. 5A based on the input bitstream 134. In the example of FIG. 5A, the next audio encoded frame that is being processed at step 305 is frame 48. (Chen Col. 7 lines 37-55).

Further, Chen teaches well known techniques in playback processing of skipping a current frame and the output being muted (whether or not the current frame contains an error therein), otherwise, the current frame is normally decoded and played. In this way, the number of transition times from normal play to mute and from mute to normal play (unmute) is reduced. In effect, the muting strategy is extended across several non-error frames depending on the accumulated error rate so that short mutings are merged into a long muting. When the error rate is high, process 280 acts to merge together adjacent error frames (mute merging) by increasing the error recovery delay period. The amount of mute merging is adaptive and is based on the error rate. At step 345, a number of different muting operations can be performed to mute the current frame. In the preferred embodiment, a smooth muting with zeros can be applied to decline the audio signal at a given rate according to a window function and in an alternate

embodiment, a frame repeat can be performed. FIG. 6 illustrates smooth muting with zeros to reduce the "pop" sounds associated with muting. In this embodiment, an attenuation or "window" function 420 is applied to the decoded audio frame represented as signal 410 to decline its amplitude. Windowing starts at the zero-cross point. The attenuation function represents the amount of the original signal 410 allowed to exist at any given time and the remainder of the audio signal is padded (e.g., replaced) with zeros to provide a mute. Smoothing functions and muting using window functions are well known (Col. 9 lines 9-38).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Oh to incorporate receiving the encoded original audio signal and applying a window function as taught by Chen to allow for the smoothing of a signal after certain frames were removed/muted, wherein a windowing function is applied to frames when skipping or muting frames if an error occurs prior to processing (Col. 9 lines 9-38), and to allow for the guaranteed elimination of any residual signal even when frames are skipped (Chen Col. 10 lines 1-26).

Re claims 4, 9, and 14, Oh teaches the method according to claim 1 wherein the desired playback speed is a predefined default value (col. 6 lines 34-38).

Re claims 5, 10, and 15, Oh teaches the method according to claim 1 wherein the desired playback speed is a programmable value (col. 6 lines 34-38).

Re claims 16-18, Oh fails to teach the method of claim 1, wherein skipping frames at a rate according to a desired playback speed further comprises skipping frames at a rate according to a desired playback speed, wherein the frames correspond to time intervals (Col. 1 lines 33-45, every other frame at a higher play back speed);.

Chen teaches error reduction of encoded frames, wherein Chen teaches error entries of error array 370 can be computed and stored by the parser process 270 of the decoder 200. There are several ways in which the AC3 data can indicate that errors are contained within a frame of encoded data. In one method, the decoder 200 can be informed of the error frame by the transport system which delivers the data. The data integrity can also be checked using the embedded CRC 220 fields for each encoded frame. Methods for using the CRC fields of an encoded frame for error detection are well known. Also, well known consistency checks on the received bitstream 134 can also be used to indicate that errors are present in a particular encoded frame. It is appreciated that at step 305 of FIG. 4, any of a number of well known processes can be used for generating the error array 370 of FIG. 5A based on the input bitstream 134. In the example of FIG. 5A, the next audio encoded frame that is being processed at step 305 is frame 48. (Chen Col. 7 lines 37-55).

Further, Chen teaches well known techniques in playback processing of skipping a current frame and the output being muted (whether or not the current frame contains an error therein), otherwise, the current frame is normally decoded and played. In this way, the number of transition times from normal play to mute and from mute to normal play (unmute) is reduced. In effect, the muting strategy is extended across several non-

error frames depending on the accumulated error rate so that short mutings are merged into a long muting. When the error rate is high, process 280 acts to merge together adjacent error frames (mute merging) by increasing the error recovery delay period. The amount of mute merging is adaptive and is based on the error rate. At step 345, a number of different muting operations can be performed to mute the current frame. In the preferred embodiment, a smooth muting with zeros can be applied to decline the audio signal at a given rate according to a window function and in an alternate embodiment, a frame repeat can be performed. FIG. 6 illustrates smooth muting with zeros to reduce the "pop" sounds associated with muting. In this embodiment, an attenuation or "window" function 420 is applied to the decoded audio frame represented as signal 410 to decline its amplitude. Windowing starts at the zero-cross point. The attenuation function represents the amount of the original signal 410 allowed to exist at any given time and the remainder of the audio signal is padded (e.g., replaced) with zeros to provide a mute. Smoothing functions and muting using window functions are well known (Col. 9 lines 9-38).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Oh to incorporate skipping frames at a rate according to a desired playback speed further comprises skipping frames at a rate according to a desired playback speed, wherein the frames correspond to time intervals as taught by Chen to allow for the smoothing of a signal after certain frames were removed/muted, wherein a windowing function is applied to frames when skipping or muting frames if an error occurs prior to processing (Col. 9 lines 9-38).

3. Claims 2, 3, 7, 8, 12, and 13 are rejected under 35 U.S.C. 103(a) as being unpatentable over Oh et al. US 5781696 (hereinafter Oh) in view of Chen et al. US 6915263 B1 (hereinafter Chen) and further in view of Kizuki et al. US 5684829 A (hereinafter Kizuki).

Re claims 2, 7, and 12, Oh in view of Chen fails to teach the method according to claim 1 wherein the encoded original audio signal is encoded in the frequency domain using one of a plurality of encoding schemes, the method further comprising frequency-domain decoding of the encoded original audio signal.

Kizuki teaches a signal encoding and decoding system such as the signal decoding system shown in FIG. 3, the bit stream received at the decoding system input is a digital audio signal represented in the frequency domain. This input is supplied to inverse quantizer-decoder 4, where it is decoded. The output of inverse quantizer-decoder 4 is fed to inverse discrete transform processor 5, where its inverse discrete transform is returned to the time domain; i.e. the inverse discrete cosine transform (IDCT), inverse discrete Fourier transform (IDFT), or inverse Karhunen-Loeve transform (IKLT), etc., as applicable, is transformed. The output of inverse discrete transform processor 5 is inverse-windowed by frame buffer 6, and output as a decoded digital audio signal represented in the time domain. The inverse windowing process multiplies each frame of the signal by the inverse of the function used to window it, thereby restoring the amplitude of the audio signal to its original state removing the window components (Kizuki Col. 2 lines 17-34).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Oh in view of Chen to incorporate audio signal is encoded in the frequency domain using one of a plurality of encoding schemes, the method further comprising frequency-domain decoding of the encoded original audio signal as taught by Kizuki to allow for an accurate method of getting information back and forth from the frequency/time domain, wherein window components can be removed and the signal content preserved in the original format (Kizuki Col. 2 lines 17-34).

Re claims 3, 8, and 13, Oh in view of Chen fails to teach the method according to claim 2 wherein said decoding comprises:

decoding said encoded signal using a decoding scheme corresponding to said one of a plurality of encoding schemes; applying an inverse transform to the encoded audio signal;

and applying an inverse window function.

Kizuki teaches a signal encoding and decoding system such as the signal decoding system shown in FIG. 3, the bit stream received at the decoding system input is a digital audio signal represented in the frequency domain. This input is supplied to inverse quantizer-decoder 4, where it is decoded. The output of inverse quantizer-decoder 4 is fed to inverse discrete transform processor 5, where its inverse discrete transform is returned to the time domain; i.e. the inverse discrete cosine transform (IDCT), inverse discrete Fourier transform (IDFT), or inverse Karhunen-Loeve transform

(IKLT), etc., as applicable, is transformed. The output of inverse discrete transform processor 5 is inverse-windowed by frame buffer 6, and output as a decoded digital audio signal represented in the time domain. The inverse windowing process multiplies each frame of the signal by the inverse of the function used to window it, thereby restoring the amplitude of the audio signal to its original state removing the window components (Kizuki Col. 2 lines 17-34).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Oh in view of Chen to incorporate decoding said encoded signal using a decoding scheme corresponding to said one of a plurality of encoding schemes; applying an inverse transform to the encoded audio signal and applying an inverse window function as taught by Kizuki to allow for an accurate method of getting information back and forth from the frequency/time domain, wherein window components can be removed and the signal content preserved in the original format (Kizuki Col. 2 lines 17-34).

(10) Response to Argument

This is in response to the appeal brief filed 11/04/2009 appealing from the Office action mailed 06/04/2009.

Re arguments directed to Chen not teaching:

- **"skipping frames at a rate according to a desired playback speed"** (Appeal Brief pages 10-13, 15, and 17).

- **"applying a window function to the remaining frames"** (Appeal Brief pages 10-13, 15, and 17).

Examiner believes that Oh teaches **skipping frames at a rate according to a desired playback speed** as well as the application of a window function. However, Examiner believes (as stated in Final office action 06/04/2009 - page 5) that Oh does not teach applying a window function to the *remaining frames*. Therefore Appellant's arguments with respect to Chen not teaching **skipping frames at a rate according to a desired playback speed** should instead be directed to Oh not teaching **skipping frames at a rate according to a desired playback speed**.

NOTE: Examiner finds the argument that "Chen does not disclose skipping frames" (Appeal brief page 10) to be irrelevant given the teachings of Oh's frame skipping as demonstrated above. Chen was not incorporated to teach **"skipping frames at a rate according to a desired playback speed"**. Even though the language of Chen states "the current frame is skipped and the output is muted" (Chen Col. 9 lines 9-39), this brief mention of frame skipping is not relevant to a *playback speed*. Therefore, motivation for using Chen resides in the applicability of a window function, which is in fact a muting operation as will be described below.

Initially Examiner would also like to point out the applicability of Oh relevant to **skipping frames at a rate according to a desired playback speed**, wherein

"in the example with skipping every other frame, is effectively twice the speed at which the original audio was but the pitch remains the same, *since the playback*

frequency remains unchanged. Hence, achieving a faster audio playback without affecting the pitch" (present invention spec. [0040]),

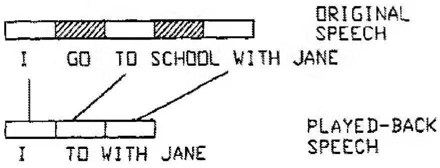
Oh teaches that the overall operation of this system is constituted by procedures of analyzing an input audio signal to vary the play-back speed while still maintaining the tone color or frequency of the audio signal (Oh Col. 4 line 43-51). This implies the same method used to skip frames.

Therefore, Oh is clearly within the scope of the present invention relative to **skipping frames at a rate according to a desired playback speed**, as Examiner believes that Oh alone explicitly teaches skipping frames at a rate based on a playback speed, wherein Oh teaches conventional uses of skipping portions in an audio signal for the purposes of speeding up play back. Oh teaches conventional methods for example, if the phrase, "I go to school with Jane", is played back at a double speed using the above-mentioned conventional method, components of the original audio respectively corresponding to the shaded portions shown in FIG. 1 are eliminated, so that only the speech "I to with Jane" is played back (Oh Col. 1 lines 33-45.

Consider that Oh improves the conventional variable speech playback method by implementing a window function, such as through the use of signal modulation by applying a window function, which provides a certain signal length extending from the position of each speech source. This procedure produces a smooth audio signal even when a signal modulation has been made by a deletion or addition of speech sources

by a speech synthesis that will be described hereinafter (Oh Col. 5 line 55 – Col. 6 line 16).

Further, please consider the Figure below extracted from Oh Figure 1 showing a conventional approach to varying the playback speed of speech.



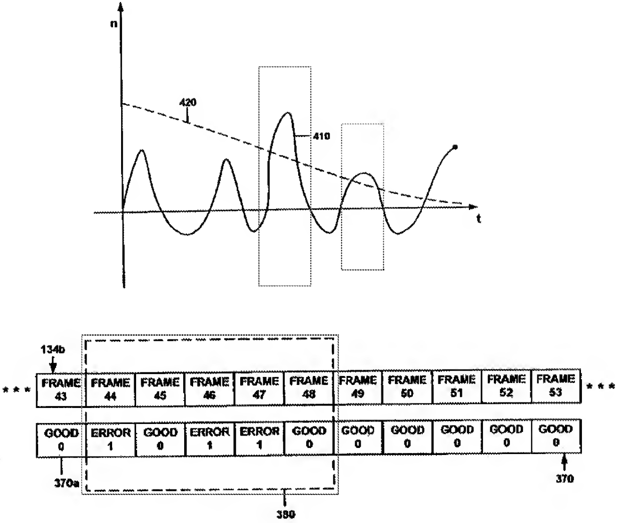
Examiner believes that Oh alone may not be sufficient and has therefore incorporated Chen. More importantly, Chen has been introduced to render obvious at what point during an algorithm a windowing function should be applied. For instance, as recited in the previous Final office action dated 06/04/2009, Examiner states that Oh fails to teach "Applying a window function *to the remaining frames*" (Final office action 06/04/2009 - page 5).

Oh teaches the well known use of window function, but Chen applies it AFTER the frames are skipped. Chen's method of skipping frames is not directed to a desired playback speed. The importance of Chen resides in the application of the window function to remaining frames. For instance, Chen teaches the use of error checking, wherein if an error is greater than the threshold, then the current frame is skipped and the output is muted (whether or not the current frame contains an error therein),

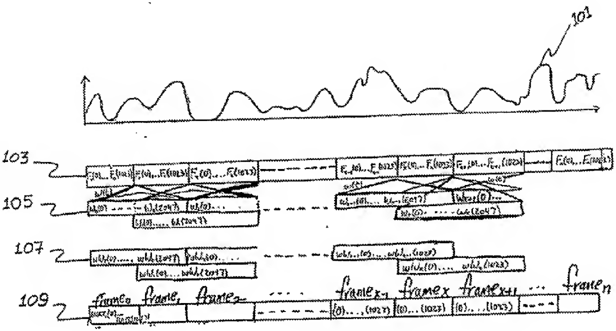
otherwise, the current frame is normally decoded and played. In this way, the number of transition times from normal play to mute and from mute to normal play (unmute) is reduced. In the preferred embodiment, a smooth muting with zeros can be applied to decline the audio signal at a given rate according to a window function and in an alternate embodiment, a frame repeat can be performed. Smoothing functions and muting using window functions are well known (Chen Col. 9 lines 9-39).

Examiner does however agree with Applicant with regard to *muting* being different from skipping frames. For instance, please consider that Oh explicitly teaches skipping speech segments as pointed out above. However Chen teaches that "muting" is a form of a window function (as is well known), that is "the output is muted" (Chen Col. 9 lines 9-39). Therefore, Oh's conventional playback concept via frame skipping is improved through the use of a window function of a frame. That is a frame is skipped (via Oh), and the output will be sent to a window function (i.e. muted via Chen). Examiner finds that Oh's variable speed capability is improved alone by Chen's window function (i.e. muting) *after* one or more frames are skipped according to a playback speed (via Oh).

Below figures 6 of Chen is shown having a red box indicating where a left over frame is to have a window function applied. Also, in figure 5B below, the frames enclosed with the dashed line are the frames that have a window function applied. Figure 5B is essentially a frame representation of the speech signal in Figure 6.



Further, figure 4 of the *present invention (Drawings)* below, shows a similar speech signal with portions skipped, wherein for example element 101 is will have frames skipped. That is, the frame(s) at this point in the audio signal is/are skipped.



(11) Related Proceeding(s) Appendix

No decision rendered by a court or the Board is identified by the examiner in the Related Appeals and Interferences section of this examiner's answer.

For the above reasons, it is believed that the rejections should be sustained.

Respectfully submitted,

/Michael C Colucci/
Examiner, Art Unit 2626

/Richemond Dorvil/
Supervisory Patent Examiner, Art Unit 2626

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